a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voiced sound mode and an unvoiced sound mode on a [the] basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination circuit section [indicates a predetermined mode], and searches combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to output a combination of a code vector and shift amount which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

2. (Amended) A speech-coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on a [the] basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination section [indicates a predetermined mode], and outputs a code vector that minimizes distortion relative to input speech bygenerating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section

3. (Amended) A speech coding apparatus including at least:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a <u>voice sound mode and an unvoiced sound</u> mode on the basis of a past quantized gain of an adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses <u>based on</u> [when] an output from said discrimination section [indicates a predetermined mode], and a gain codebook for quantizing gains, and searches combinations of code vectors stored in said codebook, a plurality of shift amounts used to shift positions of the pulses, and gain code vectors stored in said gain codebook so as to output a combination of a code vector, shift amount, and gain code vector which minimizes distortion relative to input speech; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

4. (Amended) A speech coding apparatus including at least<u>:</u>

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal, and

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal, comprising:

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook;

a sound source quantization section which has a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when an] output from said discrimination section [indicates a predetermined mode], and a gain codebook for quantizing gains, and outputs a combination of a code vector and gain code vector which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule; and

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section.

5. (Amended) A speech decoding apparatus comprising:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information;

a mode discrimination section for discriminating a voice sound mode and an unvoiced sound mode by using a past quantized gain in said adaptive codebook; and

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information based on [when] an output from said discrimination section [indicates a predetermined mode],

wherein a speech signal is reproduced by passing the sound source signal through a synthesis filter section which includes [constituted by] spectrum parameters.

6. (Amended) A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses when an output from said discrimination section indicates a predetermined mode,

Sub B1

said sound source quantization section searching combinations of code vectors stored in said codebook and a plurality of shift amounts used to shift positions of the pulses so as to poutout a combination of a code vector and shift amouint which minizes distortion relative to input speech, and further including:

a muliplexer section for outputting a combination of an ouptut from said sepectrum parameter calculation section, an output from said adaptive codeboook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,

a sound source signal reconstructing section for reconstructing a sound source signal by generating non-zero pulses from the quantized sound source information when an output from said discrimination indicates a predetermined mode, and

a synthesis filter section which is constituted by spectrum parameters and reproduces a speech signal by filtering the sound source signal.

Sub B2

7. (Amended) A speech coding/decoding apparatus comprising:

a speech coding apparatus including:

a spectrum parameter calculation section for receiving a speech signal, obtaining a spectrum parameter, and quantizing the spectrum parameter,

an adaptive codebook section for obtaining a delay and a gain from a past quantized sound source signal by using an adaptive codebook, and obtaining a residue by predicting a speech signal,

a sound source quantization section for quantizing a sound source signal of the speech signal by using the spectrum parameter and outputting the sound source signal,

a discrimination section for discriminating a voice sound mode and an unvoiced sound mode on the basis of a past quantized gain of a adaptive codebook, and

a codebook for representing a sound source signal by a combination of a plurality of non-zero pulses and collectively quantizing amplitudes or polarities of the pulses based on [when] an output from said discrimination section [indicates a predetermined mode],

said sound source quantization section for outputting a combination of a code vector and shift amount which minimizes distortion relative to input speech by generating positions of the pulses according to a predetermined rule, and further including

a multiplexer section for outputting a combination of an output from said spectrum parameter calculation section, an output from said adaptive codebook section, and an output from said sound source quantization section; and

a speech decoding apparatus including at least:

a demultiplexer section for receiving and demultiplexing a spectrum parameter, a delay of an adaptive codebook, a quantized gain, and quantized sound source information,

a mode discrimination section for discriminating a mode by using a past quantized gain in said adaptive codebook,